

Digital comm tutorial

Sociology, Communication



The process of quantisation introduces an error or noise component into the quantised signal. Derive an equation for the mean-squared quantisation error in terms of the quantization interval 'a'. ii) Hence show that the peak signal-to-quantisation noise ratio (SQNR) is $SQNR = (6n + 4.8) \text{ dB}$ Where 2^n is the number of quantisation levels. b)i) Linear quantisation is used prior to binary PCM encoding of an analogue baseband signal which has a uniform probability density function. The signal-to-quantisation noise ratio must be no less than 35 dB.

How many binary bits are required to code each quantisation level? ii) If the bit rate is 104 bits per second, what should be the maximum bandwidth of the analogue signal prior to sampling? Q2. a)i) Explain how nonlinear quantisation can be used to reduce the number of levels required to quantise a signal. ii) Explain why logarithmic quantisation is preferred. iii) What types of signal is most suitable to be processed by non-linear quantisation? b) Sketch the A-law companding curve. Explain why companding is used in voice transmission systems. c) Show that the dynamic range of the logarithmic portion of the A-law compander is 38. dB and that the improvement in signal to quantisation noise ratio realized for small signals, compared with linear quantisation, is 24 dB.

d) For an 8-bit A-law companded PCM system, calculate the SQNR obtainable and the PCM bit rate. Assume the sampling frequency is 8 KHz. Q3. a) Explain (qualitatively) how Differential Pulse Code Modulation (DPCM) can reduce the transmission bandwidth required. b) Explain what is delta modulation. Why it is particularly suited to speech signals? c) For an input sinusoid of frequency 1 kHz, estimate and compare the signal-to-error ratios

of a linear PCM coder using a sampling rate of 2. kHz and 7 bits per sample quantisation with a single-integration delta modulator producing the same gross bit rate. BASEBAND REGENERATOR / ERROR PROBABILITY / LINE CODE

Q4. a) Digital transmission systems provide better received signal quality compare to analogue transmission systems when implementing a long distance communication link. Explain briefly why this is so. b) A PCM transmission link employed 8 bit coding and uses baseband regenerator as repeater. Determine the Signal-to-Noise ratio obtainable at the receiver assuming no bit error occurred. c) An analogue transmission system required amplifiers to be spaced every 2 km apart.

Assume the Signal-to-Noise ratio of the amplifier is 65 dB, determine the maximum distance of the link before the quality of the received signal is lower than the PCM link above. d) A RF binary PSK system operates with phase states separated by 180° . The bit rate is 2.0 Mbit/s and the noise power spectral density at the input to an ideal matched filter detector is 1.0 pW/Hz. If the transmission loss between transmitter and detector is 40 dB, what power must be transmitted to achieve a probability of bit error of 10^{-6} ? For binary PSK, $P_e = ? [1 - \text{erf}(E_b/N_0)^{1/2}]$ Error function tables are provided. Q5.) Draw a simplified block diagram of a PCM regenerative repeater. b) An ideal 18 - section, copper cable, PCM link employs unipolar, NRZ, rectangular pulses on each section and a center point detection process at each repeater. The probability of error versus SNR for this transmission and detection scheme is given by [pic] If all sections were identical, and operated with a section SNR of precisely 18 dB, what would be the overall probability of error for the entire link? Q6.

(a) Sketch the typical, long term, spectrum of a speech waveform. Show on your sketch the bandwidth normally considered sufficient for telephone quality transmission. b) i) If the voice signal in part (a) is to be transmitted using 8-bit PCM and use the bandwidth upper frequency limit shown on your sketch to find the required PCM bit rate. ii) What channel bandwidth, in principle, would be required if the PCM bits were to be transmitted as perfectly rectangular pulses without distortions? iii) What is the minimum theoretical bandwidth which would allow the PCM bits to be transmitted independently (i. e. without inter-symbol interference (ISI) at the receiver sampling instants)? Explain your answer. (c) i) What is the main functions of line codes? i) The bit stream shown in Fig. Q7 is to be line-coded using the high-density substitution technique HDB3. Sketch a version of the resulting coded signal.

What are the features of HDB3 which makes it an attractive line code? [pic] Fig. Q6 TDM / PDH / SDH Q7. a)i) Describe, with the aid of a diagram, the way in which analogue telephone channels plus signalling and service information are combined in a plesiochronous time-division multiplexed system to form the primary multiplex group. ii) What sampling rate would be appropriate for each telephone channel and what would be the gross bit rate of the multiplex group?)i) Show how primary multiplex groups may be combined to form higher level multiplexes and to provide access for wideband signals. ii) Explain why it is necessary in a high order Plesiochronous digital hierarchy (PDH) to de-multiplex down to the lowest order whenever a single channel is to be extracted or inserted.

c) Calculate the number of telephone channels which can be accommodated at level 4 of a PDH. d) In the PDH, explain why the bit rate at a given level is not exactly an integer multiple of the bit rate at the level below. Q8. a) Explain why bit justification is required in a PDH network, and describe how it may be performed.) i) Determine the minimum and maximum input channel rates accommodated by an CEPT2 multiplexer. ii) Determine the rate of CEPT1 misframes caused by erroneous interpretation of a stuffed bit. Assume channel bit error rate P_e is 10^{-6} . CEPT2 parameters: Bit rate 8.448 Mbit/s Master frame length 848 bits Message length/channel 205 bits Framing bits 12 bits Stuffing control bits 12 bits Stuff bits 4 bits c)i) Explain what is frame slip. ii) In a PDH network, the primary multiplex clock generators have frequency stability of 1 part in 10⁷. Calculate the average number of frames slips per hours in a connection of 5 inter-exchange links.

Q9. a) Describe the essential features of the Plesiochronous Digital Hierarchy (PDH). b) Plesiochronous networks have a number of disadvantages by comparisons with the Synchronous Digital Hierarchy. State and explain two of them. c) Draw a block diagram illustrating the SDH. Show on your diagram the nominal STMbit rates associated with each SDH level. d)i) Describe the SDH primary-rate frame structure with particular reference to the location within the frame of the section overheads, the (administrative unit) pointers and the STM-1 payload. ii) What are the main functions of pointer?

SIGNALLING

Q10. a)i) Explain the need for signaling in a telecommunication system. ii) List the minimum basic signaling requirements, and show how they may be obtained in the subscriber loop of a typical telephone network. b) Draw a

simple block diagram illustrating the essential difference between channel-associated signaling (CAS) and common channel signaling (CCS). c) List the advantages of CCS over CAS. d) Modern digital switching systems using Stored Program Control (SPC) employ CCS. Draw a block diagram showing how CCS may be implemented. e) What is the disadvantage of CCS and how are they overcome? Q11.) Show how the ITU-T (formerly CCITT) signaling systems No. 7 conform to the International Standard Organisation, Open Systems Interconnection (ISO-OSI) model. b) What are the three types of signal units employ by the ITU-T SS No. 7? What is the function of each? c)i) How is the channel associated signaling handled by the 30+2 PCM primary multiplex frame? ii) Calculate the bit rate of the signaling channel with one voice channel. TELETRAFFIC THEORY Q12. a) In a switching system for which blocked calls are lost, the average number of calls per hour is 200 with an average holding time of 3 minutes.

Estimate the number of trunks required to achieve a grade of service of 0. 1 %. b) On the average during the busy hour, traffic generated in exchange A and exchange B is shown in table Q13. Assume no tandem traffic, estimate the number of trunk channels (two way connections) required for a grade service of 1 %: i) if the same lines are used for incoming and outgoing calls, ii) if separate lines are used for incoming and outgoing calls. Evaluate the above options and propose a cost effective solution. What is the minimum number of trunk lines required to serve the two exchanges? | Exchange A | Exchange B | | Exchange A |- | 36 Erlang | | Exchange B | 43 Erlang |- | Table Q13 c) Calculate the number of channels needed in a seven-cell re-use pattern cellular systems to achieve a blocking probability of 1 % if there are

2800 calls per cell per hour, each of average duration of 1.8 minutes. (use traffic table). Q13.

a) Define traffic intensity and congestion.) Explain why it is necessary to determine the traffic variations as a function of time for a telephone exchange. c) For a telephone exchange designed based on blocked call lost assumption, the probability of there being k calls in progress with N trunks carrying traffic A Erlang is given by: $P_k = \frac{A^k}{k!} P_0$ i) Explain what is meant by blocked call lost. Give an evaluation the effect of this assumption. ii) Derive an equation for the probability all servers are busy and the subscriber encountered call blocking. State the assumptions made for the above equation to be valid.) A PBX with 250 internal lines has 10 trunks to the public network. i) What is the probability of call blocking if each internal line is involved in four external calls with an average duration of 2.5 minutes per call, per eight-hour working day? ii) How many additional trunk connections would be required to improve the grade-of service to better than 0.5 %?

Q14. a) In a queueing system, the average rate of packet transmission is μ frames per second, and the average arrival rate of data is λ packets per second. The probability that there will be n packets in the queue is

$P_n = (1 - \rho) \rho^n$ where $\rho = \lambda / \mu$ b) Derive an equation for the average number of packets in the queue and show how this varies with the parameter ρ . c) How would you use this equation to design the node in a packet-switched system? d) If the switching node has a transmission capacity of 800 packets per second and the packet arrival rate is 500 packets per second. i) Calculate the average number of packets in the queue and hence ii) Calculate the average waiting time per packet. iii) What is the

mean delay introduced by the switching node on a packet?) A common – channel signalling system uses a 64 kbits/s data link to serve a group of 1500 speech circuits on a route between two exchanges. The busy-hour traffic is 1000 E and the average call duration is two minutes.

On average each call requires transmission of ten messages (five signals plus five responses) and the average message length is 20 octets. Calculate the percentage of messages which encounter delay and the mean delay for these messages. DATA COMMUNICATION NETWORK Q15. a) i) Describe the principle of data communication by packet switching. ii) Evaluate the advantages of this strategy by comparison with circuit switching.) Show how the format of a packet can allow inclusion of routing, error correction, synchronisation and data. c) A packet switch has a single outgoing link at 2.048 Mbit/s. The average length of each packet is 960 bytes. If the average packet delay through the switch must be less than 20 ms, assuming an M/M/1 queue, determine the i) maximum total packet arrival rate ii) average length of the queue. Q16a) Outline the ISO-OSI data communication network model. b) i) At which layer of the ISO-OSI model does the routing information provided? ii) Name and describe briefly two common routing protocols for the Wide Area Network (WAN).

ii) Compare the relative performance of the protocols. iv) give an example of the connection standard applicable to each. c) Describe the format of a High-level Data Link Control (HDLC) packet and describe how this could be employed to implement call set-up, data transfer and call clearing in a virtual circuit. ISDN / B-ISDN Q17. a) Most national tele-traffic networks have evolved from systems using analogue telephony and signaling and

electromechanical switching. Show, using diagrams, how it has been possible to develop Integrated Digital Networks (IDN) whilst retaining much of the transmission network.) An IDN is required to provide communication of information in addition to digital telephony signals. Describe the other signals necessary to operate an IDN and show how these can be integrated within a single network. c) Outline the potential benefits of an ISDN. d) Describe the data handling capabilities of Basic Rate Access and Primary Rate Access ISDN services. What are the gross bit rates in each case? Q18. a) Describe the process which takes place in a packet speech transmission system and outline the transmission delay which might be expected.) In a packet communication network packets arrive at a switch according to a Poisson distribution with a mean arrival rate of 4 packet/s.

The service time is exponentially distributed with a mean value of 100 ms. Assuming that each packet contains 70 bytes and the output transmission rate is 5.6 kbit/s. How long, on average, does a packet have to wait in the queue? If the switch in part (c) is limited in length to 10 packets, what is the probability of losing packets? c) What extensions to these access processes will be required to handle multi-media terminals and what data transfer method will be most appropriate?) What are the numerical values of the following: i) ATM cell size. ii) ATM information field size. iii) SDH STM-1 bit rate. iv) PCM voice channel bit rate. e) Use your answer in part (d) to find the expected total network delay (including packetisation delay) experienced by a voice signal transmitted over an ATM network connection operating at the SDH STM-1 bit rate. The connection traverses 8 switching centers, each of which introduces a mean delay equal to 98 ATM cells. The transmission path

length is 350 km in total, and the specific delay of the transmission medium is 5 (s/km).